Abstract of the Disclosure

Systems and methods for converting a data stream from a first sample rate to a second sample rate using a sample rate converter that employs selectable filters. In one embodiment, the filters are implemented by providing multiple sets of filter coefficients in a memory, selecting one of the sets of filter coefficients and performing coefficient interpolation to produce filter coefficients that are convolved with the input data stream to produce a re-sampled output data stream. The input signal can be an audio signal that is convolved with interpolated polyphase filter coefficients in the sample rate converter of a digital PWM audio amplifier. The set of filter coefficients can be selected by a value stored in a filter selection register that is modifiable by a DSP or by user input. The sets of filter coefficients can be stored in a single memory and interpolated according to a cubic spline interpolation algorithm.